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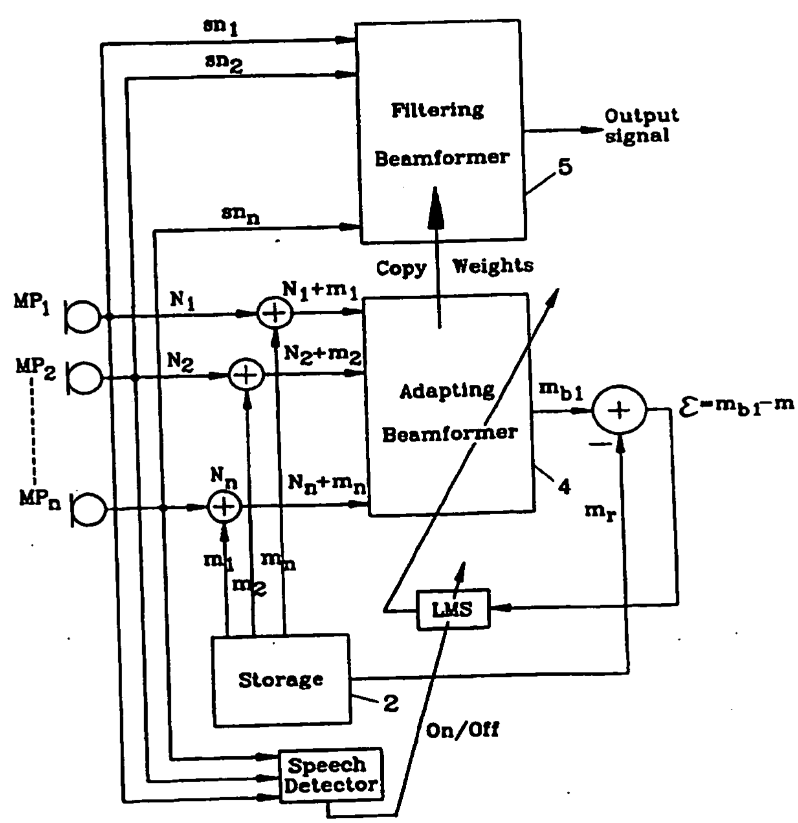
(54) Title: ADAPTIVE MICROPHONE ARRANGEMENT AND METHOD FOR ADAPTING TO AN INCOMING TARGET-NOISE SIGNAL

(57) Abstract

The invention relates to an adaptive microphone arrangement with one or more microphones (MP_1, ..., MP_n) comprising a signal detecting arrangement for detecting target input signals, a signal forming arrangement and signals storing means. The input signals comprise a calibration signal (m_1, ..., m_n) and a second noise signal (N_1, ..., N_n) wherein the calibration input signal is recorded and stored in a storing means (2). The signal forming arrangement comprises a first signal forming means (4) and a second signal forming means (5) wherein the first signal forming means (4) comprises adapting means for treating the sum of the calibration signal and a noise signal to provide filtering coefficients which then are copied to and used in the second signal forming means (5) on the target-noise input signal and wherein the adapting signals and the target-noise signals are input under essentially the same conditions.



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ADAPTIVE MICROPHONE ARRANGEMENT AND METHOD FOR ADAPTING TO AN INCOMING TARGET-NOISE SIGNAL

FIELD OF THE INVENTION

The present invention relates to an adaptive microphone arrangement as referred to in the first part of claim 1. The invention furthermore relates to a method for adapting to an incoming target signal.

The conditions under which a microphone arrangement is to be used vary to a great extent. Sometimes the environment is very noisy, as for example in a car or any moving vehicle or similar, moreover also in workshops, storehouses etc. When so called hands-free operation is applied, the requirements on the microphone arrangement is even more demanding among others due to the distance from the source of the speech or whatever it may, be to the microphones. E.g. the noisy environment in a car severely degrades the performance of so called hands free mobile telephones and speech recognition devices.

STATE OF THE ART

A number of attempts have been done to improve the quality of e.g. speech signals in noisy environments. These solutions are generally based on spectral subtraction, Wiener-filtering and array technics. Thus, through the use of speech generating models and various algorithms, more and more a priori information about speech signals is used in speech recognition and speech coding arrangements. In order to ensure that the a priori information is correct, an acceptable signal-to-noise ratio is required which in turn implies a need for noise reduction under various adverse conditions such as e.g. hands-free operation of telephones or speech recognition in cars. With the abovementioned solutions the signal-to-noise ratio has been increased. For example methods or arrangements based on special filtering adaptive microphone arrays. However, in all these known applications the near field considerations are of great

importance. This in turn leads to problems which are hard to describe in detailed manner with an a priori model. For example a beamformer with inherent theoretical modelling representing the acoustic field in a car will with considerable probability provide an array which is based on a priori information which partly is incorrect. In known arrangements (such as e.g. "Methods for noise reduction applied to speech input systems" by K. Krochel in Proc. VLSI and Computer Peripherals VLSI and Microelectronic Applications in Intelligent peripherals and their Interconnection, Network", 8-12 May 1989 p. 2/82-87, automatical modulation is used to provide the wanted signals. This does however not work if the a priori information is wrong or incorrect and furthermore requires advanced mathematical models etc. Moreover in a number of known arrangement or methods so called noise cancellers are used and they are built on reduction principles etc. which are complicated and generally it does not give a very good result but leads to complicated arrangements etc.

SUMMARY OF THE INVENTION

It is an object of the present invention to provide an adaptive microphone arrangement as initially referred to which gives a good signal-to-noise ratio. It is furthermore an object of the invention to provide an arrangement wherein no mathematical modelling is required about signal statistics, aconstic field, microphone array geometry and characteristics of the electronic equipment etc. further object with invention is to provide an arrangement which can be used for so called hands-free operation in cars etc. and which provides good output signals, e.g. a signal, having a good signal-to-noise ratio. Still another object of invention is to provide a microphone arrangement which is insensitive to channel mismatch.
Still another object with invention is to provide an arrangement which is robust and easy to use. Further objects with the invention is to provide an arrangement wherein a calibration signal or a reference signal is easily obtained and which is,

to the greatest extent possible, correct. It is also an object of the invention to provide an arrangement wherein the discrepancy between the filtering functions during speech recognition training and operation is small when used with speech recognition devices.

A further object of the invention is to provide a method for adapting to an incoming target signal.
These as well as other objects are achieved through an arrangement having the characteristics of the characterizing part of claim 1.
The objects are moreover achieved through a method having the characteristics of claim 20.
A number of advantageous embodiments are given by the appended subclaims. Advantageously the signal forming arrangement comprises an adaptive beamformer and a filtering beamformer. In a particularly advantageous embodiment the calibration signal is a speech signal or even more particularly a typical speech signal or a signal with a speech influenced spectrum. Most particularly the calibration signal is recorded on site, i.e. it is recorded using the same equipment and in an advantageous embodiment at the same location as when the target target-noise signal is produced. Preferably the storage comprises a digital storage, or even more particularly one digital storage for each input calibration signal, each for a separate microphone. The calibration signal may comprise a number of (secondary) calibration signals. i.e. calibration signals from each microphone which are combined into a so called desired signal. For adaptation of the sum of the calibration signal and the noise signal which according to an advantageous embodiment comprises pure noise, the adapting means uses an adaptive algorithm which e.g. may be the so called LMS (Least Mean Square) algorithm or some other algorithm, for example the RLS (Recursive Least Square) or any other appropriate algorithm. Particularly either one of the calibration signals or a combination of two or more thereof often is used as a so called desired signal in the algorithm means with which the sum of the

calibration signal and the noise signal is compared in a manner known per se. During adaptation, during which no target signal or no speech is provided, a number of filtering coefficients are obtained in the adaptive beamformer in a manner known per se. The filtering coefficients are copied to and used in the second beamformer or the filtering beamformer. When a target (target-noise) signal is input, or a speaker or similar is active, the adaptation of the adaptive beamformer is switched off and no adaptation takes place. Then the target signal or e.g. the speech signal is filtered through the filtering beamformer. Generally the first and second beamformers comprise filters such as e.g. FIR-filters (Finite Impulse Response), the adaptation coefficients thereof being optimized adaptively to the actual noise level or noise situation and to the equipment "on site".

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will in the following be further described in a non-limiting way under reference to the accompanying drawings wherein:

- Fig. 1 illustrates a calibration phase and
- Fig. 2 illustrates an adaptive filtering phase.

DETAILED DESCRIPTION OF THE INVENTION

In the following an embodiment will be described wherein an array of microphones is arranged for example in a car. In the Figs. an array comprising n microphones (MP_1, MP_2,..., MP_n) is illustrated wherein n can be any number from one upwards and is chosen depending on the actual circumstances and the relevant environment. Thus there may be either one or more microphones. In one particular embodiment 8 microphone are used but this of course merely constitutes an example. The microphones may be of any appropriate quality or of any kind. If however they are of a standard quality, they generally have a con-

siderable spread in performance which in turn poses high demands on the beamformer as to easily incorporate a calibration step. According to the invention training sequences are recorded from different positions in the environment of e.g. a true speaker position in a real situation with the actual system and with no noise present. The training sequences or the calibration signals are then gathered into a storage and later used as so called training signals in the adaptive phase. Therethrough an inherent calibration signal is obtained and it is generally possible to wheigh interesting frequency bands and spatial points. The arrangement according to the invention is accurate for the actual situation and it does not depend on the geometry of the array of microphones or similarities between elements or on calibration or matching of amplifiers or other electronic equipment etc. The microphone arrangement generally uses two sets of input data, namely the target-noise signals in a filtering beamformer and the recorded calibrations signals plus the noise signals in the adaptive beamformer. In the first and second beamformer respectively, i.e. the adaptive beamformer and the filtering beamformer respectively, the signals are filtered with so called FIR-filters or Finite Impulse Response filters or a so called tapped delay line, which carries out a linear combination of input data.

In the described embodiment, the microphone arrangement may particularly be used for so called hands free operation.

The microphone arrangement according to Fig. 1 comprises a number of microphones MP_1, MP_2, ..., MP_n wherein the microphones are arranged and placed in any desired manner. The input calibration signals M_1, ..., M_n undergo, an anti - aliasing operation and an A/D conversion in a conversion block 1 whereafter the signals, now designated m_1, m_2, ..., m_n are recorded in a calibration signal storage 2. The calibration signals m_1, ..., m_n are also used in the adaptive means as will be further described later on. The calibration signal is to be provided as a pure calibration signal, i.e. in the case of a car or similar there should generally be no noise upon their generation and recording, i.e. the car should be parked with no motor

on etc. Then a typical speech signal or signal with a speech influenced spectrum, from the typical speaker position, is recorded in the calibration signal storage 2. This is preferably a digital storage or more particularly a number of digital storages, each for one microphone channel. These recorded signals form calibration signals m_1, ..., m_n. The adaptive means or the adaptive beamformer 4 can advantageously be calibrated on-site in a car or similar e.g. by using either a loudspeaker or letting the speaker read a representative sequence. The sequences received in each microphone channel are gathered into the calibration signal storage 2. This means that the channels from the speaker or the loudspeaker or similar to A/D converters are included. As already mentioned above, the environmental noise level should be as low as possible in order to obtain a good signal-to-noise ratio in a desired signal which may be either one of the input calibration signals m_1, ..., m_{n1}, or a combination of two or more of the calibration signals m_1, m_2, ..., m_n. In a preferred embodiment the situation as well as the equipment is generally time-invariant wherethrough the microphone arrangement has been provided with calibration signals which can be combined to form the desired signals as referred to above. Further, as already discussed above, the separate microphones MP_1, MP_2, ..., MP_n and their placement can be chosen in any appropriate manner. According to a preferred embodiment, for the obtaining of a robust system, the speaker position or the loudspeaker position is changed in such way that it is moved around and in the vicinity of the speakers normal position during the recording of the calibration signal into the storage. The recorded calibration signals from different positions are according to a preferred embodiment superimposed to provide weighted average training signals or calibration signals or reference signals. As already referred to above, these signals are gathered into the storage-2. As can be seen from Fig. 2, those signals, m_1, m_2, m_n and m_r which forms so called calibration signals, or reference signals, are then used as well as training signals as, e.g. in a combined form, as a desired signal or reference signal for use

during adaptation.

After the calibration phase, wherein the calibration signals are recorded and stored in the storage, an adaptive phase follows. During this phase there is no calibration input signal. The situation is very generally a noisy situation, in the case of the car it may relate to a situation wherein the speaker is silent and wherein the car is moving, i.e. the motor is running etc. The input signals to the adapting beamformer 4 are formed by the sum of the in the storage 2 stored calibration signals m_1, m_2, ...,m_n and the noise signals N_1, N_2, ..., N_n respectively. Thus the speaker or similar is silent. The storage also comprises an arrangement (not shown) wherein e.g. a combined desired signal m_r is formed. This might also be formed by one of the input calibration signals m_1, ..., m_n or a combination of just some of them. To the adaptive beamformer 4 is introduced stored speech signals m_1, ..., m_n plus noise N_1, ..., N_n. A known reference signal or a desired signal m_r which has passed through the same electronic equipment when no noise was present is also obtained. The adaptive filters of the adapting beamformer 4 therethrough are provided with all the information that is needed to adapt to the correct filter coefficients e.g. in the least square sense or applying the LMS-algorithm (or any other appropriate algorithm). In a manner known per se the reference signal or the desired signal m_r is subtracted from the output signal m_{b1} from the adaptive beamformer 4 and the difference ϵ is formed etc. Thus the signals from a "typical" speaker and the real speaker originate from the same acoustical environment and meet the same electronic equipment etc. Therefore the adaptive microphone arrangement will be calibrated "on site" to the prevailing acoustic environment and to the placement of the microphones etc. as well as to the individual properties of the microphones, amplifiers, A/D - converters and so on.

When the coefficients of the digital filters of the adaptive beamformer 4 has been optimized adaptively to the current noise situation and to the actual equipment, these are copied to the

second beamformer or the filtering beamformer 5. The filtering beamformer 5 operates when the speaker or similar is active. When the speaker (or similar) is active, the adaptation is switched off, either automatically or manually e.g. by a "push-to-talk"-function. This relates to a preferred embodiment; it is however not necessary. If the adaptation is switched off, however, this is done to avoid echo-effects and/or to provide a more robust system so that the adaptive filters cannot operate on the real speech signal. The target signal or the speech signal, comprising speech plus noise, sn_1, sn_2, ..., sn_3 is merely filtered through the filtering beamformer 5. During the filtering in the filtering beamformer 5, the filtering coefficients are fixed and the output signal is obtained from the filtering beamformer 5. According to a preferred embodiment, as soon as the speaker stops to speak, the adaptation in the adapting beamformer is continued. The filtering beamformer preferably works continuously and without any calibration signal.

The different components of the microphone arrangement can be of any desired kind. A number of different known microphone types can be used. Different filters can also be used of which so called FIR-filters merely constitute one example. Also the storage can be chosen in any appropriate way. The sampling frequency may likewise take a number of different values.

The invention may also in a number of other aspect be varied in a number of different ways merely being limited by the scope of the claims.

CLAIMS

1. Adaptive microphone arrangement with at least one microphone (MP_1, MP_2, ..., MP_n) wherein the arrangement comprises a signal detecting arrangement for detecting target input signals, a signal forming arrangement and signal storing means, characterized in that, the input signals comprise a calibration signal (m_1, ..., m_n) a noise signal (N_1, ..., N_n) and a target-noise signal (sn_1, ..., sn_n) wherein the calibration input (m_1, ..., m_n) signal is recorded and stored in the storing means (2) and in that the signal forming arrangement comprises a first signal forming means (4) and a second signal forming means (5) wherein the first signal forming means (4) comprises adapting means treating the sum of the calibration signal (m_1, ..., m_n) from the storage (2) and the noise signal (N_1, ..., N_n) thereby providing filtering coefficients and in that the filtering coefficients obtained from the first signal forming means (4) are used in the second signal forming means (5) on the target input signal and in that the calibration and noise signals and the target-noise signals are input under essentially the same conditions.

2. Arrangement according to claim 1, characterized in that, the adapting means (4) is an adaptive beamformer.

3. Arrangement according to claim 2, characterized in that, the second signal forming means (5) is a filtering beamformer.

4. Arrangement according to anyone of the preceding claims, characterized in that, the noise signal (N_1, ..., N_n) is a pure noise signal.

5. Arrangement according to anyone of claims 1 - 4, characterized in that,

the calibration signal (m_1, ..., m_n) is a speech signal.

6. Arrangement according to claim 5, characterized in that, the calibration signal (m_1, ..., m_n) is a typical speech signal or speech-spectrum influenced-signal.

7. Arrangement according to anyone of the preceding claims, characterized in that, the calibration signal (m_1, ..., m_n) is recorded on site.

8. Arrangement according to anyone of the preceding claims, characterized in that, the storage (2) comprises at least one digital storage.

9. Arrangement according to claim 8, characterized in that, the input calibration signal comprises a number of calibration signals (m_1, ..., m_n) which are combined into a desired signal (m_r).

10. Arrangement according to anyone of the preceding claims, characterized in that, the adapting means (4) uses an adaptive algorithm (LSM, RMS) for the adaptation.

11. Arrangement according to claim 10, characterized in that, the adaptive algorithm is the so called LMS-algorithm or some other gradient algorithm or similar.

12. Arrangement according to claim 11, characterized in that, one of the calibration signals (M_1, ..., M_n) or a combination thereof is used as a so called desired signal in the adaptive algorithm.

13. Arrangement according to anyone of the preceding claims, c h a r a c t e r i z e d i n t h a t, an essentially pure noise signal (N_1, ..., N_n) is combined with the essentially pure calibration signal (m_1, ..., m_n) in the first adaptive beamformer (4) wherein the adaption coefficients are so adapted that the output signal from the adapting beamformer (4) is similar to a combination of the calibration signals.

14. Arrangement according to anyone of the preceding claims, c h a r a c t e r i z e d i n t h a t, the filtering coefficients obtained from the adaptation in the adaptive beamformer (4) are copied to, and used in, the filtering beamformer (5).

15. Arrangement according to anyone of the preceding claims, c h a r a c t e r i z e d i n t h a t, the adaptation by the adaptive beamformer (4) is switched off when a speaker or similar is active, i.e. up on input of the target-noise signal.

16. Arrangement according to anyone of the preceding claims, c h a r a c t e r i z e d i n t h a t, the target signal or the speech signal merely is filtered through the filtering beamformer (5).

17. Arrangement according to anyone of the preceding claims, c h a r a c t e r i z e d i n t h a t, it comprises one single microphone (MP_1).

18. Arrangement according to anyone of claims 1-16, c h a r a c t e r i z e d i n t h a t, it comprises an array of microphones (MP_1, ..., MP_n).

19. Arrangement according to anyone of the preceding claims, c h a r a c t e r i z e d i n t h a t, the first and second beamformers (4,5) comprise filters such

as e.g. FIR-filter (Finite Impulse Response Filters) and in that the adaptive coefficients thereof are optimized adaptively to the actual noise level or noise situation and to the equipment on site.

20. Method for adapting to an incoming target-noise signal comprising the steps of:

- recording at least one calibration signal coming essentially from the same physical position and meeting essentially the same equipment as a target-noise signal,

- storing the calibration signal (s) in the digital storage,

- providing an adaptive beamformer with the sum of a pure noise signal and the calibration signal(s) and one of or a combination of the calibration signals as desired signal for the adapting process,

- carrying out an adaption in the adaptive beamformer providing adapting coefficients such that the adaptive beamformer suppresses surrounding noise and listens to a calibration signal,

- copying the adapting coefficients from the adaptive beamformer to a filtering beamformer,

- providing the filtering beamformer with a target-noise signal during the provision of which the adaptation is switched off,

- providing an output signal from the filtering beamformer.

21. Method according to claim 20 wherein the filtering beamformer works without a calibration signal in a continuous manner.

1:2

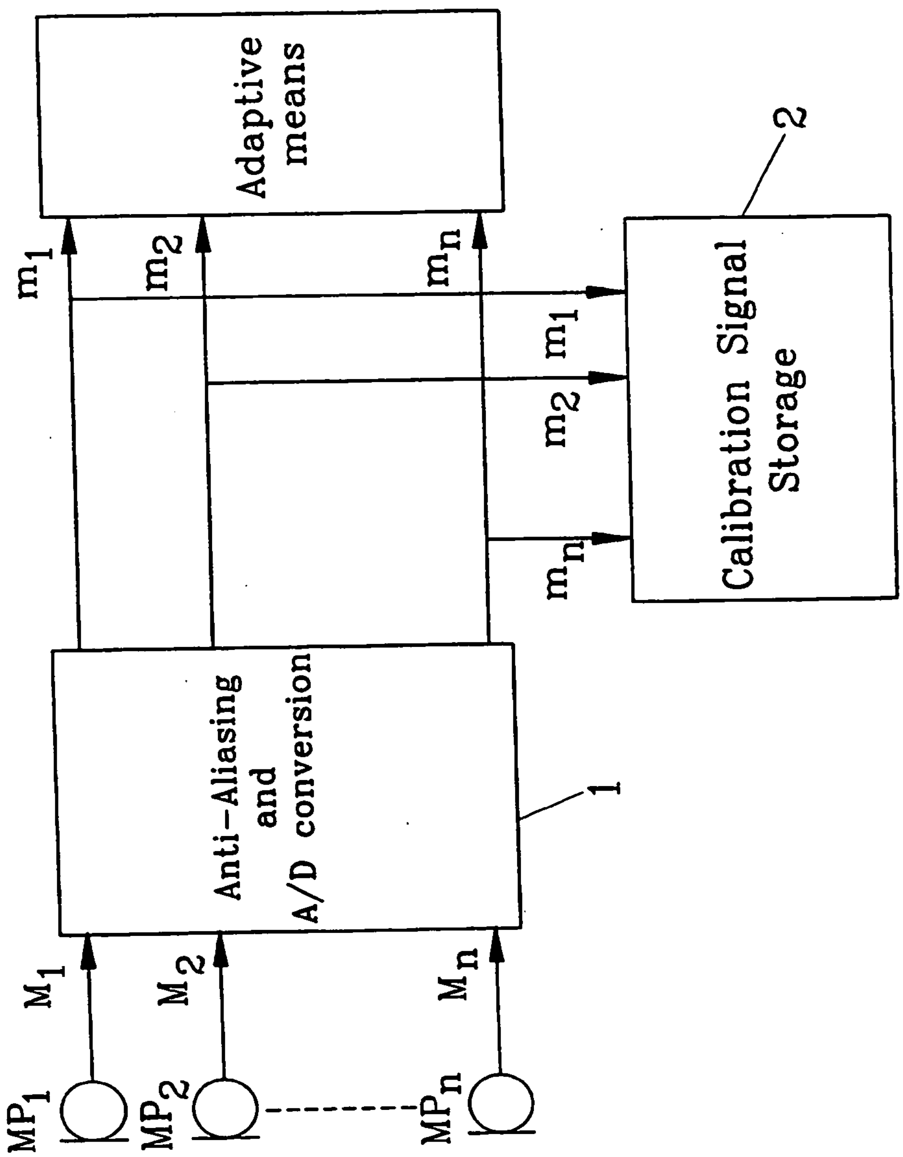
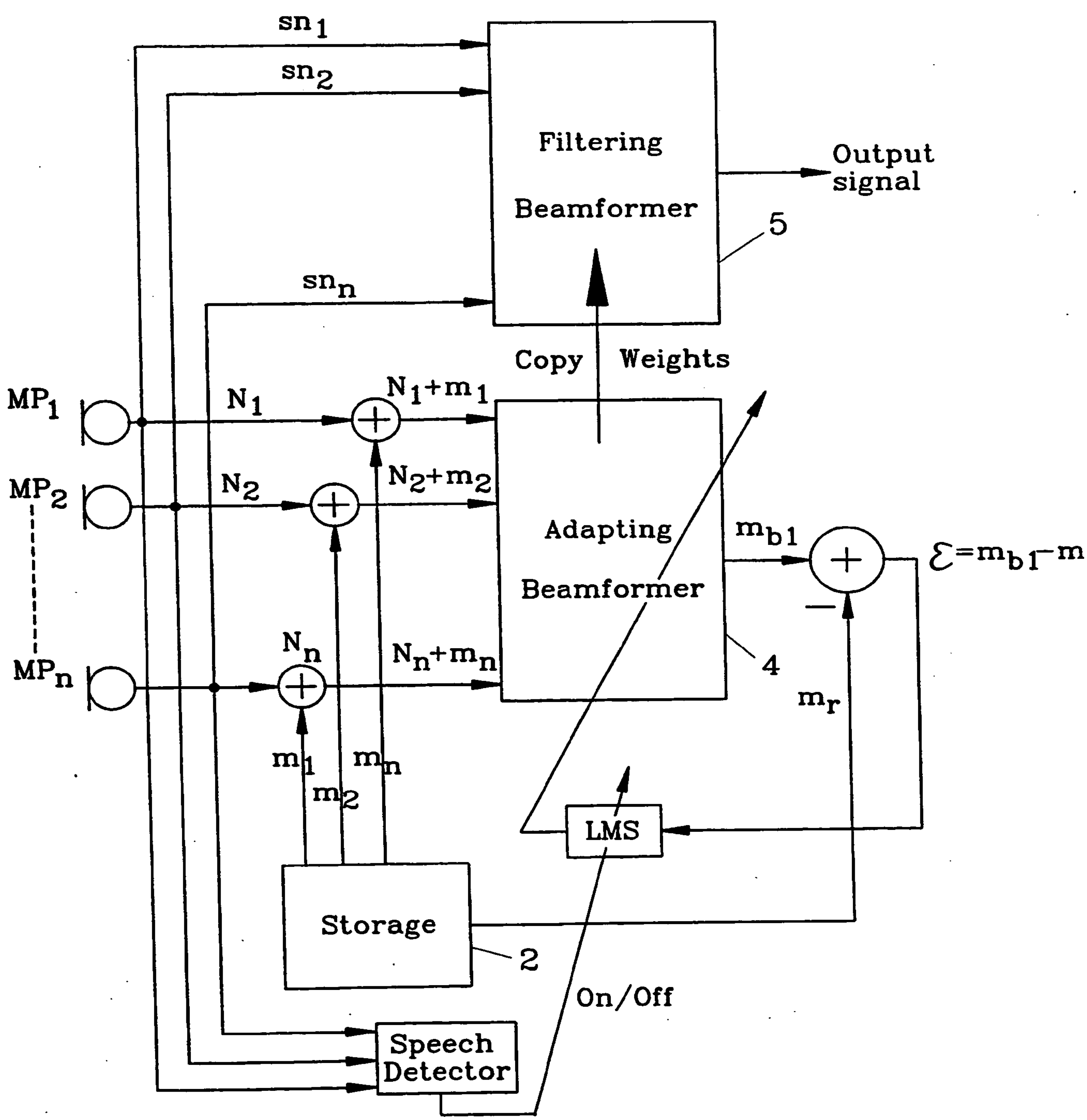

Adaptive means
Calibration Signal Storage
Anti-Aliasing and A/D conversion
m_1
m_2
m_n
M_1
M_2
M_n
MP_1
MP_2
MP_n
1
2

FIG. 1

FIG.2


sn1
sn2
snn
Filtering
Beamformer
Output signal
5
Copy Weights
MP1
MP2
MPn
N1
N2
Nn
N1+m1
N2+m2
Nn+mn
Adapting
Beamformer
4
mb1
ℰ=mb1-m
mr
m1
m2
mn
LMS
Storage
2
On/Off
Speech Detector

INTERNATIONAL SEARCH REPORT

International application No.
PCT/SE 95/00718

A. CLASSIFICATION OF SUBJECT MATTER

IPC6: H04M 1/19, G10L 3/02
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC6: H04M, G10L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

SE,DK,FI,NO classes as above

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

CLAIMS, WPI, INSPEC

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 4956867 A (PATRICK M. ZUREK ET AL), 11 Sept 1990 (11.09.90), column 3, line 51 - column 5, line 47 --	1-21
A	Speech Communication, Volume 9, 1990, (North-Holland), Dirk VAN COMPERNOLLE et al, "SPEECH RECOGNITION IN NOISY ENVIRONMENTS WITH THE AID OF MICROPHONE ARRAYS" page 433 - page 442 --	1-21
A	IEEE TRANSACTION ON VEHICULAR TECHNOLOGY, Volume 42, No 4, November 1993, Sven Nordholm et al, "Adaptive Array Noise Suppression of Handsfree Speaker Input in Cars" page 514 - page 518 -- --------	1-21

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INTERNATIONAL SEARCH REPORT

Information on patent family members

28/08/95

International application No.
PCT/SE 95/00718

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
US-A- 4956867	11/09/90	WO-A- 9013215	01/11/90

Form PCT/ISA/210 (patent family annex) (July 1992)